An adaptive coded modulation with multi-levels QoS analysis in multimedia environment

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Abstract: This article proposes an adaptive coded modulation (ACM) with multi-levels qualities of service analysis, which can be used to perform physical layer functions in software radio receivers for beyond third generation systems. ACM approche employs multiple coding and modulation schemes in order to adapt instantaneously spectral efficiency to the signal-to-noise ratio variations SNR. The idea is to study transmission schemes performance which allow dynamic bandwidth sharing among various services in multimedia mobile and wireless networks in different channel environments. The aim is to select the appropriate modulation and channel coding modes based on signal quality measurement via a decision feedback block (DFB). Decision taken by the DFB block is based on different QoS parameters that depend in class and type of multimedia traffic. The DBF block is introduced in order to mitigate the channel quality fluctuations. The channel coded adaptive modulation schemes can be viewed as a lower complexity alternative of mitigating the channel quality fluctuations in comparison to multiple-transmitter and multiple-receiver. Implementing DFB necessitate to simulate and analyze different combinations between modulation and channel coding blocs for different data types.

Keywords and phrases: Channel coding, adaptive modulation, Signaling, image, Multimedia, SDR.

1 Introduction

The channel coding and transmission schemes conception dedicated to a particular application is based on a complex set of contradictory design factors [1].

An adaptive physical layer schemes is the transmitter action in response to time varying channel quality which is related to many factors [2, 3, 4, 5, and 7]. A reliable estimation of the channel transfer quality is required for the next active transmit time slot so the transmitter has to select the appropriate modulation and channel coding modes. Based on signal quality measurements, the receiver must inform the transmitter via signaling messages. On the other hand, the receiver must to be informed as to which reception scheme to employ for the received packet. The feedback channel is assumed to be error-free and instantaneous (see figure 1).

Transmission parameters can be adapted to the channel conditions. For example, adjusting the modulation levels number in response to the anticipated SNR. The adaptive channel coding parameters introduces code rate, adaptive interleaving and puncturing for convolution and turbo codes, or varying block lengths for block codes. These techniques can be combined with adaptive modulation mode selection, which is the principle our study objective.

Signaling plays an important role in adaptive systems. There are three signaling modes in adaptive systems: open loop modulation signaling mode, closed loop modulation signaling mode and blind modulation mode detection at the receiver [8].

If the channel quality estimation and parameter adaptation have been performed at the transmitter of a particular link, based on open-loop adaptation, then the resulting set of parameters has to be communicated to the receiver in order to successfully demodulate and decode. If the receiver itself determines the requested parameter set to be used by the remote transmitter this is the closed loop scenario, then the same amount of information has to be transported to the remote transmitter in the reverse link. Blind detection algorithms estimate the employed modulation mode directly from the received data symbols, therefore avoiding the loss of data capacity due to signaling. There are three algorithms for this blind detection, which are Euclidean distance, MSE and Kullback-Leibler distance [9]. Our solution described by fig.1 is based on the closed loop modulation-signaling mode.

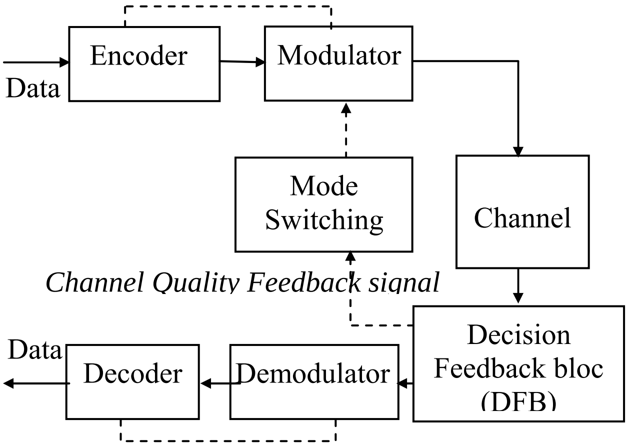


Fig.1: Adaptive physical layer scheme

This article is structured in six sections. The first section is the introduction, which describes the approach of an adaptive coded modulation context and modulation signaling modes aspects in adaptive systems. Section 2 zooms on the ACM concepts chosen for different mobile and wireless networks. Section 3 describes QoS enhancements in mobile and wireless network. Performance analysis of different coding modulation schemes in multimedia environment and simulation results are examined in section 4. An outline for our DBF and mode switching algorithms with multimedia constraint are presented in section 5 and finally the conclusions of this article are given in section 6.

2. ACM in mobile and wireless network:

ACM offer a significant improvement for new wireless and mobile generation. HSDPA, EGPRS, GPRS-136, 802.11a/g and WiMAX are standardization's that use ACM techniques.

HSDPA, which refers to, the improvement made in the UMTS downlink, introduces a number of new technical capabilities to the radio access network. The use of Adaptive Coding Modulation is one of these capabilities [8].

ACM is the fundamental technology that allows HSDPA to surpass the data rates of its predecessors. Traditionally, systems that utilize CDMA have used a constant modulation scheme (usually M-PSK), with fast power control to adapt to changes in channel conditions. Instead, ACM transmits with a constant power while the Modulation and Coding Scheme is altered to adapt to these variations. Table 1 shows the different throughput rates achieved based on the modulation, the coding rate, and the number of HS-DSCH (High Speed - Downlink Shared Channels) codes in use.

Modulation	Coding rate	Through-put (5codes) Mbps	Through-put (10codes) Mbps	Through-put (15codes) Mbps
QPSK	1/4	0.6	1.2	1.8
	2/4	1.2	2.4	3.6
	3/4	1.8	3.6	5.4
16 QAM	2/4	2.4	4.8	7.2
	3/4	3.6	7.2	10.7
	4/4	4.8	9.6	14.4

Table 1: HSDPA Throughput Rates

EGPRS [9, 10] is based on nine completely new channel coding methods. The first four Modulation and Coding Scheme, MCS1-MCS4, are GMSK modulated and the remaining five MCS5 to MCS9 are 8PSK modulated. In addition, EGPRS utilises an ACM scheme allowing dynamic selection depending on local channel conditions such as weak reception or noise. The dynamic modulation and coding scheme selection is Link Adaptation (LA) [9, 11, and 12]. Table 2 [10, 13] shows EGPRS modulation and coding schemes with their maximum throughputs.

	Modulation Method	1 TS (Kbps)	4 TS's (Kbps)	8 TS's (Kbps)
MCS-1	GMSK	8.8	35.2	70.4
MCS-2	GMSK	11.2	44.8	89.6
MCS-3	GMSK	14.8	59.2	118.4
MCS-4	GMSK	17.6	70.4	140.8
MCS-5	8-PSK	22.4	89.6	179.2
MCS-6	8-PSK	29.6	118.4	236.8
MCS-7	8-PSK	44.8	179.2	358.4
MCS-8	8-PSK	54.4	217.6	435.2
MCS-9	8-PSK	59.2	236.8	473.6

Table2: EGPRS modulation and coding schemes

GPRS-136 [14] uses an adaptive modulation, which is provided via a switching between constellation sizes (M) of 4, 8 and 16 as a function of the channel quality between the base station and mobile terminal in a given cell [15].

Standards such as WiFi and WiMax operate using a large number of modulations and coding modes. Each mode offers a Packet Error Rate (PER) versus Signal to Noise Ratio (SNR) performance. Generally, higher-rate modes require a greater SNR and thus operate with reduced range. In all current WLAN standards, link adaptation is designed to maximize the error-free throughput. When transmitting video, data throughput is not the only metric of importance; issues such as latency and jitter also must be considered. Often, the mode that maximizes the error-free data throughput will produce excessive transmission delay and/or jitter. For data networks, such delays are rarely of concern. However, for real-time applications (voice and video broadcasting) link adaptation module is very important. When considering the WLANs at 2.4 GHz and 5 GHz Physical layer, such as IEEE 802.11g and IEEE 802.11a respectively, numerous modes are available, each providing different throughput and reliability levels. Table 3 summarizes the different operating modes available for the IEEE 802.11a/g PHY layer. They range from BPSK 1/2 rate (Mode 1), which provides a nominal bit rate of 6 Mbps to 64 QAM 3/4 rate (Mode 7), with a nominal bit rate of 54 Mbps. The BPSK 1/2 rate mode provides a more reliable transmission link than the 64 QAM 3/4 rate mode for a given receive power.

Modulation	Rendement	Rendement net au dessus de PHY	Octet per symbole
BPSK	1/2	6Mbit/s	3
BPSK	3/4	9 Mbit/s	4.5
QPSK	1/2	12 Mbit/s	6
QPSK	3/4	18 Mbit/s	9
16-QAM	9/16	27 Mbit/s	13.5
16-QAM	3/4	36 Mbit/s	18
Option			
64-QAM	3/4	54 Mbit/s	27

Table 3: Mode dependent parameters for IEEE 802.11a/g.

The Wireless-MAN (WiMax) air interface provides system designers with four different modulation schemes BPSK (used for signaling messages), QPSK, 16-QAM and 64-QAM. It also provides multiple coding schemes derived from combinations of GF (256) Reed-Solomon (RS) codes and rate 2/3 convolution codes. For very high levels of robustness, turbo codes are also included. This enables system designers to define suitable modes using the optimal modulation and FEC pairs for different channel conditions. The standard defines the necessary messaging and signaling framework that can accommodate link adaptation on both forward and reverse links.

3. QoS enhancement

The emerging use of real-time multimedia applications over wireless and mobile networks makes the Quality of Service (QoS) support a key problem [16]. Multimedia over wireless or mobile networks poses many challenges [17]. Each data type such as voice or video has different system requirements like bandwidth and tolerance to delay. The Table 4 shows the QoS requirements of different data types [18].

	Voice	Data	Image
Delay	<100ms	---	<100ms
Packet Loss	<1%	0	<4%
BER	10^{-3}	10^{-6}	10^{-6}
Data Rate	8-32 Kbps	1-100 Mbps	10 Mbps
Traffic	Continuous	Bursty	---

Table 4: Multimedia Data Requirements.

An adaptable physical layer [19, 20] enables wireless and mobile applications to proactively optimise physical layer performance using some application-defined set of metrics. For example, an adaptable physical layer may be used to switch from a carrier frequency with a low signal to noise ratio (SNR) to a carrier frequency with a much higher SNR given that the BER needs to be reduced.

The modulation scheme performance is measured by its ability to preserve the encoded data accuracy. In mobile wireless networks, path loss, fading, and interference cause variations in the received SNR. Such variations also cause variations in the BER, because the lower the SNR, the more difficult it is for the receivers to decode the signal.

This tradeoff is illustrated in section 5, which shows the BER as a function of the SNR for several different modulation schemes. Notice that for each modulation scheme the BER decreases with increasing SNR. Also, notice that for a given SNR, an increase in data rate results in an increase in BER.

To evaluate the impact of the channel on the image quality, we need to compare the received (possibly distorted) image with the actually sent image. In table 5 we evaluate PSNR to image quality. The transformation from objective scale to subjective is based on different simulations. For this mapping we have considered different QCIF images (Lena, news, Foreman). We have considered only simulation in AWGN channel (no multipath) without any coded modulation scheme. In fig.2, we illustrate the Lena QCIF image quality for different SNR.

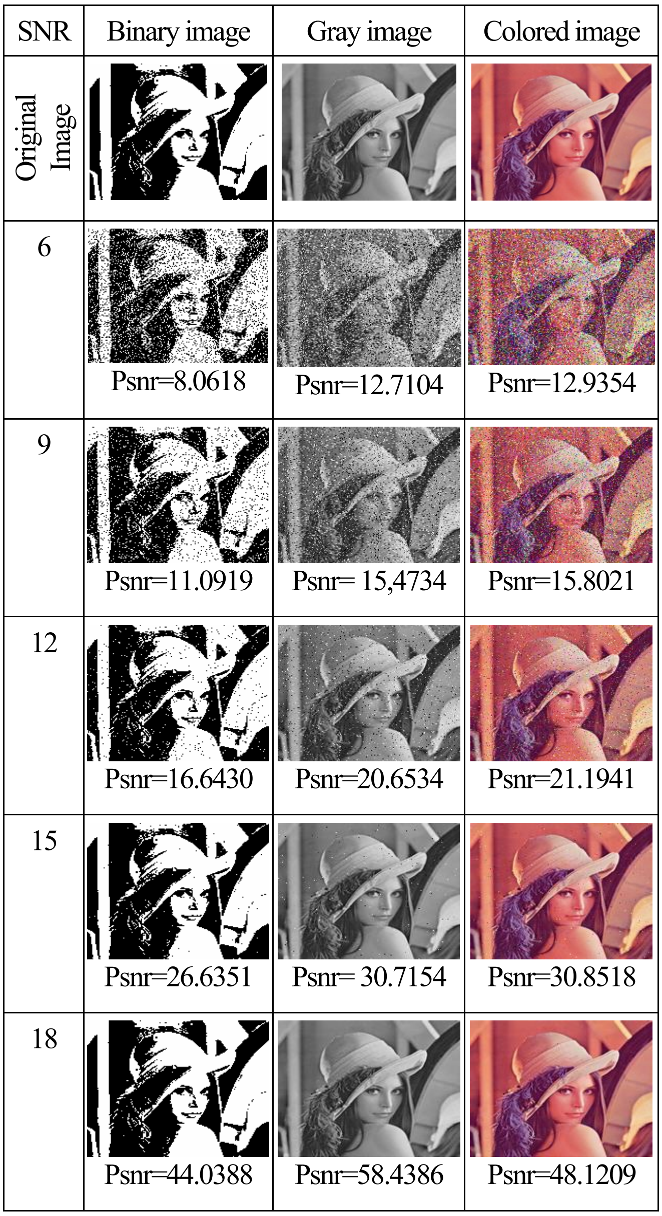


Fig.2: Lena QCIF image Quality for different SNR

The PSNR value is a number that reflects the transmitted image quality with respect to the original sequence. High PSNR values indicate a strong correlation with the original sequence and hence the transmitted image is deemed to be of good quality. The formulas used to calculate PSNR for binary or gray image are shown below in Equation (1) and (2) [21].

$$MSE=\frac{\sum_{x=0,y=0}^{W,H}[O(x,y)-R(x,y)]^2}{W*H} \quad (1)$$

$$PSNR=20\log_{10}\left(\frac{\max(S(i))-\min(S(i))}{\sqrt{MSE}}\right) \quad (2)$$

In Equation 1, O denotes the original image and R denotes the received. The images are W×H dimensional, and the MSE is calculated as the squared difference between all corresponding pixels, divided by the the image pixels number. All the images used in the experimental work of this paper have QCIF dimensions, so W is 176 and H is 144. PSNR is derived by setting the MSE to the pixels maximum value, which is equal to (max (S(i)). PSNR is a log based metric and it is measured in decibels (dB).

The formula used to calculate PSNR color image is shown below in Equation (3) [22].

$$PSNR=20\log_{10}\left(\frac{\max(S(i))-\min(S(i))}{\sqrt{MSE_{RGB}}}\right)$$
$$MSE_{RGB}=\frac{MSE_R+MSE_G+MSE_B}{3} \quad (3)$$

PSNR [dB]	Quality
> 37	5 (Excellent)
31 - 37	4 (Good)
22 - 30	3 (Fair)
15 – 22	2 (Poor)
< 15	1 (Bad)

Table 5: PSNR evaluation to image quality

Typical PSNR values range between 20dB, which corresponds to very poor quality, and 40dB, which corresponds to excellent quality that typically shows no perceptible difference from the original images.

In voice communications, the mean opinion score (MOS) provides a numerical measure of the human speech quality at the receiving end. MOS indicates the speech quality perceived by the listener and can range from 1(bad) to 5(excellent) (see table 6).

Score	MOS	DMOS	
5	excellent	inaudible	no effort required
4	good, toll quality	audible, but not annoying	no appreciable effort
3	fair	slightly annoying	moderate effort
2	poor	annoying	Considerable effort
1	bad	very annoying	no meaning

Table 6 mean opinion score (MOS) and degradation MOS.

The most popular objective measurements is SNR audio which is able to predict subjective quality with good correlation in a very wide range conditions, which include coding distortions, errors, noise, filtering, delay and variable delay. SNR audio compares the original (reference) signal with the degraded system under test output using a perceptual model. The *SNR* take into account the intensity of the reference. This is done by dividing the total reference power by the total error power. Taking the logarithm is one way of reducing values range, and the 10 in front is the convenience factor. The formula used to calculate SNR audio is shown in Equation (4).

$$SNR\,(S,S_R)=\frac{\sum_{x=1}^{N} S(x)^2}{\sum_{x=1}^{N}(S(x)-S_R(x))^2}$$

with $SNRdb = 10\log_{10} SNR(S,S_R)$ **(4)**

In table 7 we have evaluated SNR audio to sound quality. The transformation from objective scale to subjective is based on different simulations. For this mapping we have analyzed different quality obtained with different SNR illustrated in fig.3.

We have considered only simulation in AWGN channel without any coded modulation scheme. To evaluate the channel impact on the sound quality, we need to compare the received sound with the actually sent sound.

In fig. 3, the input wave sequence which is named “blip” is illustrated, in the other cases two sound sequences the first is the output sequence and the second is the error sequence are represented.

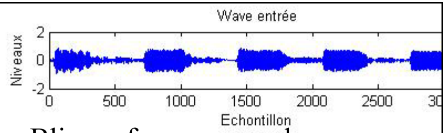
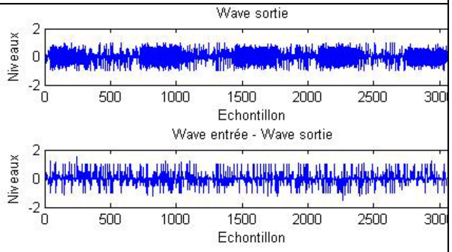
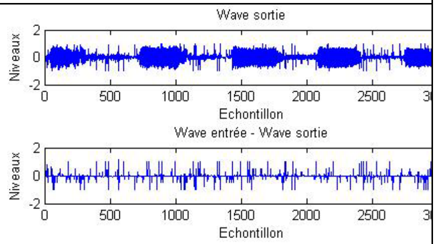
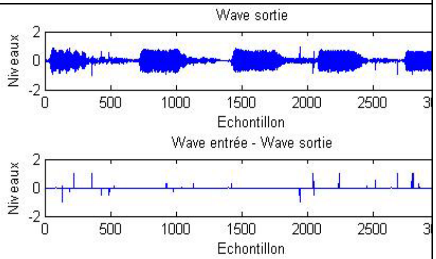
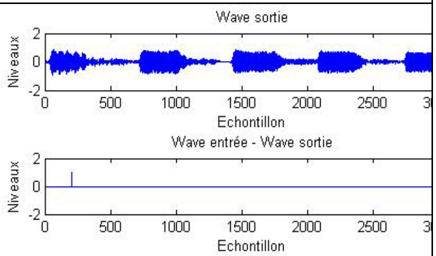
SNR chanel	
	 Blip : reference speech
9	 SNR audio=0.8650db
12	 SNR audio=6.1337db
15	 SNR audio=15.0701db
18	 SNR audio=25.8105

Figure 3: Audio quality for different SNR channel.

SNR audio [dB]	Quality
> 25	5 (Excellent)
25 - 15	4 (Good)
15 - 6	3 (Fair)
6 – 2	2 (Poor)
< 2	1 (Bad)

Table 7: SNR audio evaluation to sound quality

4. Coding modulation schemes performance analysis in multimedia environment

Simulations are used to derive Bit Error Rate values for a range of modulation efficiency and FEC Code Rate parameters. Simulations used in this section were developed using the Matlab Communications Toolbox. The communications library form this software package was adapted to create a number of different modulation and coding scheme that were investigated. In Fig.4 we have evaluated LENA QCIF image transmission quality (PSNR) with different modulation schemes used for different SNR. In Fig. 5 we have evaluated audio sequence (3168 sample: every sample is coded over one byte) transmission quality (SNR audio) with different modulation schemes used for different SNR.

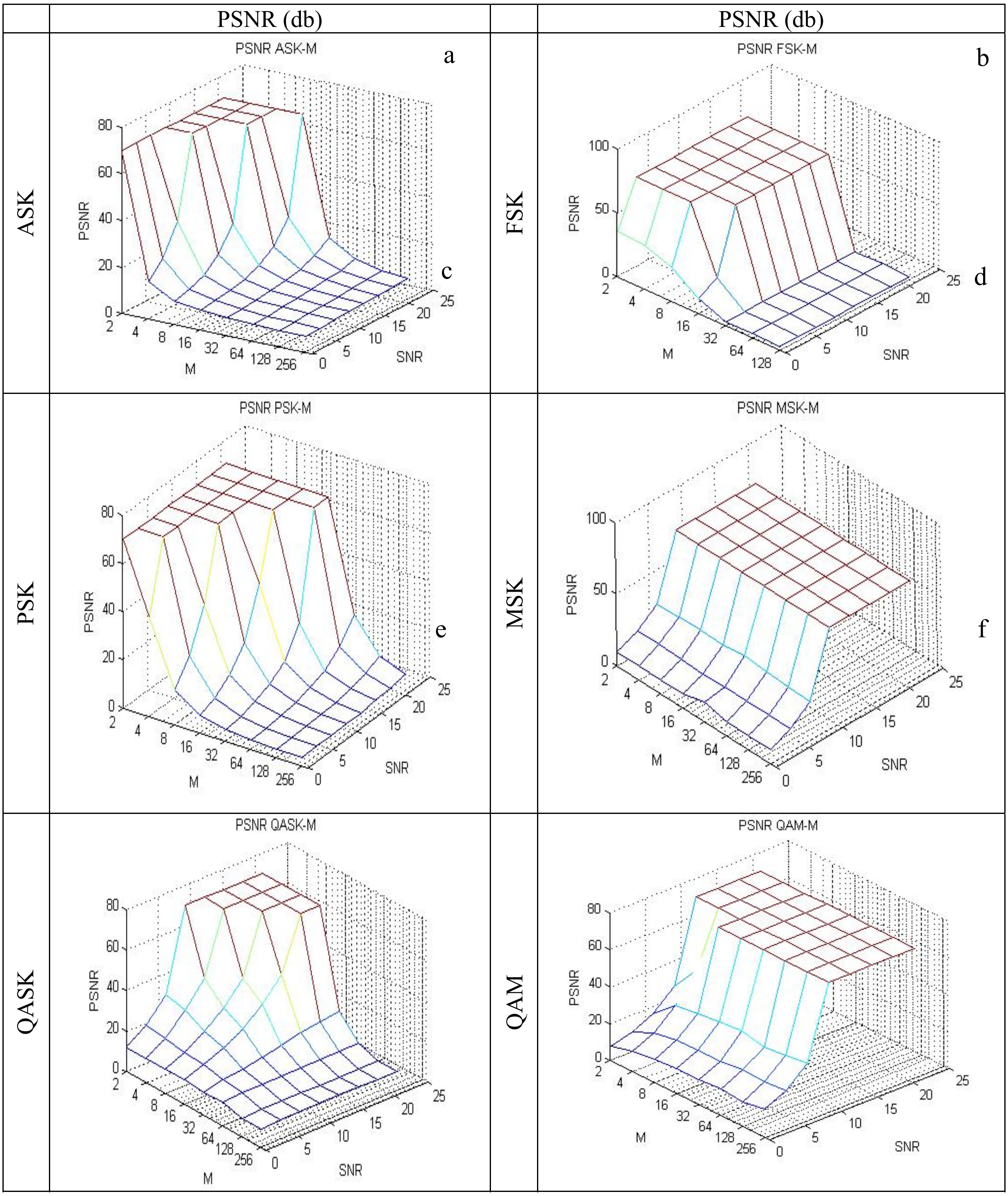


Fig.4: LENA QCIF image transmission quality with different modulation schemes.

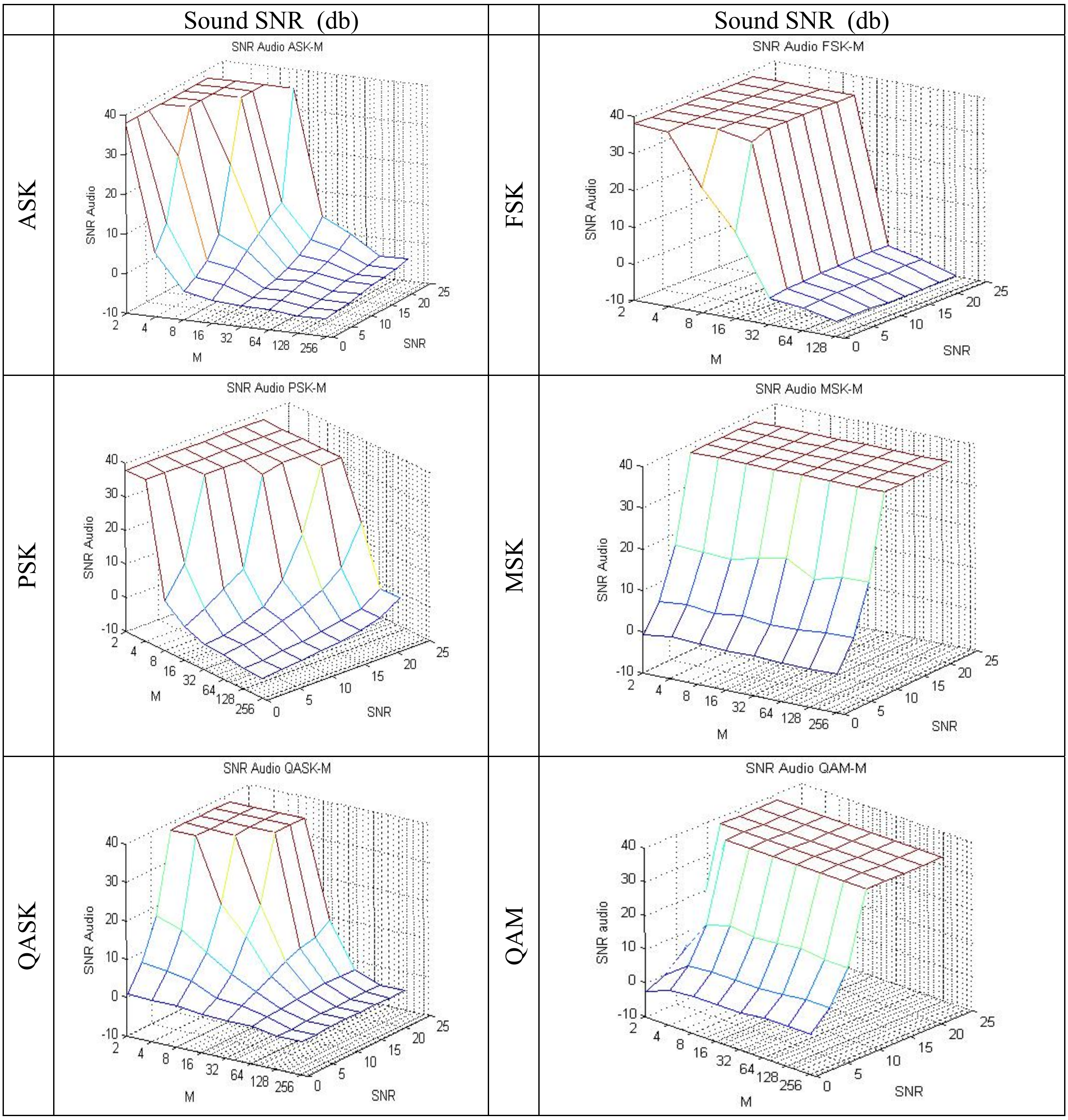


Fig.5: Audio sequence transmission quality (SNR audio) with different modulation schemes.

The transmission quality over ASK-M modem shown in fig.4.a is very decreased especially when M is greater than 2 or when the SNR ratio is near to 0.

The transmission quality over FSK-M modem is good when M is less than 16; in this case the FSK modem isn't very sensitive to the effects of the noise (SNR→ 0) when the SNR ratio is near to 0. When M is greater than 16, the transmission quality is much decreased.

The PSK-M modem has its better quality at high SNR value and low M value. When M increase and SNR decrease the quality is much reduced.

The MSK modem isn't sensitive to the M variation. The quality is good when the SNR is greater than 10 dB.

The QASK-M modem has the better transmission quality when M is low and SNR is high, over those values the degradation is very quick.

The QAM-M modem is not very sensitive to the M variation; the quality is excellent when SNR is greater than 15 dB.

Table 8 summarizes the SNR required to achieve a PSNR > 30 (good quality) for all types of modulation schemes over an AWGN channel.

	SNR required to achieve a PSNR > 30					
M	ASK-M	PSK-M	FSK-M	MSK	QASK	QAM
2	0	0	0	6.5	6	10
4	5	0	0	6.5	6.8	6.8
8	12	4	1.2	6.5	9	6.5
16	18	10	4	6.5	11	6.2
32	25	15.6	None	6.5	29.6	6.2
64	None	25	None	6.5	None	6.1
128	None	None	None	6.5	None	6.1
256	None	None	None	6.5	None	6

Table 8: SNR required to get a PSNR > 30

At a low rate (M is low: ∈{2, 4, 8}), we can choose PSK-M or FSK-M modulation schemes in order to obtain a good quality at a low SNR channel (very noise). At a medium rate (M ∈{16, 32), we can choose QASK-M or QAM-M modulation schemes in order to obtain a good quality at a medium SNR channel. At a high rate (M ∈{64,128, 256), we can only choose QAM-M modulation schemes in order to obtain a good quality at a medium SNR channel.

The performance of a modulation scheme can be measured by its robustness against path loss, interferences, and fading that cause variations in the received SNR. Such variations also cause variations in the PSNR, since the higher the SNR, the easier it is to demodulate and decode the received bits. Compared to other modulations schemes, BPSK has the maximum PSNR value for a given SNR. For this reason, it is used as the basic mode for each Physical layer since it has the maximum coverage range among all transmission modes.

We have compared the performance gain obtained by transmitting sound over different modulation schemes. Therefore, the practical aspect of the performance is based on the SNR value the current level of reliability in terms of SNR audio for each modulation can be determined. Table 9 summarizes the SNR required to achieve a sound SNR > 15 (good quality) for all types of modulation schemes over an AWGN channel.

	SNR required to achieve a sound SNR > 15					
M	ASK-M	PSK-M	FSK-M	MSK	QASK	QAM
2	0	0	0	5.5	5.8	9
4	4	0	0	6	6	6
8	10	3.5	0	6	8	5.8
16	20	10	1	6	9	5.7
32	27	14	None	6	24	5.7
64	None	20	None	6	None	5.7
128	None	None	None	6	None	5.7
256	None	None	None	6	None	5.7

Table9: SNR required to get sound SNR > 15.

The comparison between table 8 and table 9 provides that there isn't an important difference so with the same SNR and the same modulation scheme we can have the quality level (MOS). So for the DBF and ACM switching algorithms we don't distinguish between information types.

We have also compared the PSNR obtained by transmitting Lena image (Qcif) over different codec modem schemes (see fig. 6) in this simulation the SNR of the channel is fixed at 9dB and M is fixed at 16, which corresponds to a medium rate. In order to evaluate the coding module effect on the transmission schemes, in the first step we have calculated the performance of different modulation schemes without coding module and in the second step we have associated modulation module with coding module in transmission schemes.

We notice that the ASK modulation quality is very low although the use of coders (such as: RS, viterbi…), for this reason Ask modulation isn't very used in networks standards.

PSK and QASK quality can be performed with coders and decoders schemes.

QAM, MSK and FSK have good quality without coding module so when we use those modulation schemes we can omit coding module.

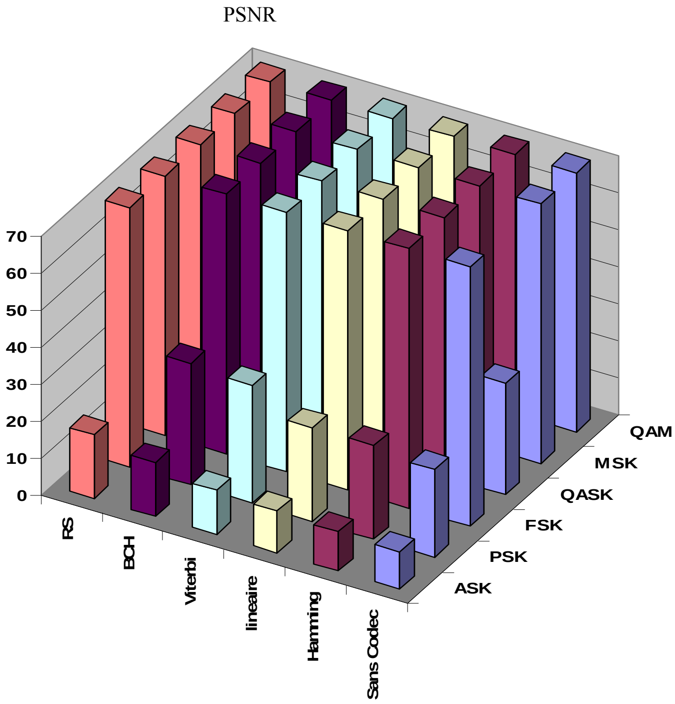


Fig. 6: LENA QCIF image transmission quality (PSNR) with different codec-modem schemes.

5. DBF and ACM switching algorithms

Simpler modulation schemes tend to have higher quality. That is, since simpler modulation schemes generally represent lower bit rates, a frame transmitted with a lower bit rate is less likely to experience errors than a frame transmitted with a higher bit rate at the same SNR.

The algorithm that we suggest to be used by the modulation coding mode selector can be described as the following:

Modulation and Coding Mode switching:

$$
\begin{array}{lll}
M_1 & \text{if} & 0 \leq SNR \leq \alpha_1 \\
M_2 & \text{if} & \alpha_1 \leq SNR \leq \alpha_2 \\
M_3 & \text{if} & \alpha_2 \leq SNR \leq \alpha_3 \\
 & \vdots & \vdots \\
M_N & \text{if} & \alpha_{N-1} \leq SNR \leq \propto
\end{array}
$$

Where Mi with i = 1,2,3…N are modulation and coding modes listed from the lowest data bits per symbol to the highest data bits per symbol, and α1, α2 , α3 ….αN-1 are the corresponding switching modes. For the modulation and Coding Mode switching: We have specified Mi and αi (see table 10) that offer PSNR >30dB and SNRson >15dB.

Débit	M	SNR (dB)	Modulation
Low	2 / 4 / 8	$0 \leq SNR \leq 6$	PSK-M ou FSK-M
		$6 < SNR$	MSK-M
		$SNR > 7$	QASK-M ou QAM-M
Medium	16	$4 \leq SNR \leq$	FSK-16
		$6 < SNR \leq 10$	FSK-16 ou QAM-16
		$SNR > 10$	QASK-16 ou PSK-16
	32	$SNR \geq 15$	QAM-32 ou PSK-32
		$SNR < 15$	QAM-32
High	64 / 128 / 256	$SNR \geq 6$	QAM-M

Table 10: Switching mode selection

6. Conclusion

The modulation technique employed has direct effects in transmission quality. We analyzed the performance of these modulation schemes in terms of PSNR used for different SNR. The results obtained reveal that the SNR ratio in the feedback channel may have a significant impact.

For the adaptive modulation scheme, a Peak Signal to Noise Ratio = 30 and also a sound SNR>15 which corresponding to a good quality are chosen in order to specify a mode switching algorithm. A total of nine modes have been specified.

As open issue, it might also be possible to take in consideration the SDR and cross layer approaches in order to develop a generic adaptive coding modulation system.

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